REMARKS

Applicant submits the priority application and drawings as originally filed and have made amendments as summarized below.

The Cross-Reference to Related Applications section has been added to identify the application as a Continuation application, which claims priority from copending U.S. application 10/105,600.

Applicant has amended the originally filed parent application by canceling claims 2, 11-13, 16, and 23-24. Claims 3-5 and 19-22 have been amended to change the claim dependencies and to eliminate subject matter previously allowed in the parent application.

No amendment is related to the statutory requirements of patentability or for the purpose of narrowing the scope of any claim within the meaning of Festo Corp. v. Shoketsu Kinzoku Kogyo Kabushiki Co. Ltd., No. 95-1066 (Fed. Cir. September 26, 2003).

No new matter has been added.

The following remarks are submitted to address rejections in the Office Action dated July 1, 2003, for the parent application for claims present in the Continuation application.

Claim Rejections - 35 USC §103

Claims 1, 6-10, 14-15, 17-18, 21, and 25 were rejected in the parent application under 35 USC §103(a) as being unpatentable over Ramanathan et al. (USPN 6,076,113, hereinafter "Ramanathan") in view of Murpy et al. (USPN 6,542,499, hereinafter "Murpy") as applied to claims above, and further in view of Jagadeesan (USPN 6,577,996, hereinafter "Jagadeesan").

Ramanathan provides a method and system for evaluating user-perceived network performance by connecting a remote terminal to a data service system and emulating a communication protocol to transfer data reliably and in sequence with congestion control. The communication protocol includes mechanisms for acknowledgment and retransmission and a dynamic window size, which is restricted to not to be greater than a predetermined maximum window size with the amount of data transferred restricted to a predetermined data transfer size.

Murphy provides a call fallback scheme in a packet switched network (PSTN). After receiving incoming calls, a Voice over IP (VoIP) link is established over a packet switched network with a destination endpoint. VoIP packets are generated from the incoming calls and sent over the VoIP link to the destination endpoint. When a low quality of service condition is detected on the VoIP link with the destination endpoint, a fallback call is established with the destination endpoint over a circuit switched network. The VoIP packets for the incoming calls are redirected from the VoIP link to the circuit switched data link. As opposed to simply hairpinning a time division multiplexed (TDM) voice call back over the PSTN network, the same VoIP packets for the incoming calls originally destine for the destination endpoint over the packet switched network are rerouted through the fallback call. This simplifies synchronization with VoIP packets sent over the VoIP network. Because VoIP packets for more than one call can be sent over the fallback call, the cost of maintaining the fallback call is also substantially reduced.

Jagadeesan provides a method and apparatus for objectively evaluating sound quality of a signal processor or transmission channel by analyzing the distortion in a series of test sound frames compared to a series of sample sound frames. Sequences of test sound frames having distortion levels that are greater than a temporal distortion threshold are detected and an average length and a maximum length of these sequences are calculated. Also, individual test sound frames having distortion levels that are greater than an outlier distortion threshold are detected and a percentage of these frames present in the series of test sound frames is calculated. The average distortion level in the series of test sound frames and a variance of the distortion level in the test sound frames are calculated. These parameters are then combined to produce a objective sound quality score which can be used to evaluate a sound transmission system or select a transmission channel for communication of sound signals.

Pertaining to claims 1, 6-10, 14-15, 17-18, 21, and 25, Applicant respectfully traverses the rejection because the independent claims contain the following limitations as exemplified in claim 1 of:

> "a modem termination system (MTS);
> a voice band tester (VBT) coupled to the MTS, the VBT being located at a first location;
> a modem tester coupled to the MTS, the modem tester being located at a second location remote from the first location, the modem tester

adapted to provide a first communication signal to the VBT via the MTS; and

a Voice over Internet Packet (VoIP) monitoring device coupled to the MTS and the VBT, the VoIP monitoring device adapted to monitor the first communication signal, and calculate a first Quality of Services (QoS) score based on traffic density between the MTS and the VBT; wherein the VBT is adapted to:

calculate a first Transmission Impairment Test (TIT) score based on the first communication signal and a first received communication signal received by the VBT from the modem tester, and

provide the first TIT score to the VoIP monitoring device."

The Examiner states in the Office Action of 7-1-03 that:

"Ramanathan et al. discloses the invention **(claims 1, 6-10, 14-15, 17-18, 21, and 25)** as claimed including a communications network, comprising:

(a) a modem termination system (MTS) (see Fig. 2, element 24, col. 4, lines 65-67)." [bold in original and underlining for clarity]

Applicant respectfully disagrees. Ramanathan element 24 is a router as shown in FIG. 2 and explained in Ramanathan col. 4, lines 50-67 (which includes lines 65-67):

"The ISS 10 includes a router 24 for routing data to and from the subscriber sites 12, 14 and 16 upon receiving a request from a subscriber/user. The router 24 functions to connect the subscriber sites 12, 14 and 16 to the appropriate on-premises servers 18, 20 and 22, or to the global Internet 30 or the other ISSs 28. The router 24 may operate in the Asynchronous Transfer Mode (ATM) to provide high bandwidth packet-based switching and multiplexing." [underlining for clarity]

It is respectfully submitted that those having ordinary skill in the art know that a modem termination system would not read on a router because signal conversion capabilities are lacking in the router. For example, www.searchNetworking.com explains that:

"The CTMS (cable Modem Termination System) sends and receives digital cable modem signals on a cable network, receiving signals sent upstream from a user's cable modem, converting the signals into IP packets and routing the signals to an Internet Service Provider for connection to the Internet. The CMTS also can send signals downstream to the user's cable modem. Cable modems cannot communicate directly with each other, they must communicate by channeling their signals through the CTMS."

Thus, the term "modem termination system" is more limiting than "router", which does not include the limitations inherent in the modem termination system.

The Examiner continues:

> "[Ramanathan et al. discloses the invention **(claims 1, 6-10, 14-15, 17-18, 21, and 25)** as claimed including a communications network, comprising:]
>
> (c) a modem tester coupled to the MTS, the modem tester being located at a second location remote from the first location, the modem tester adapted to provide a first communication signal to the VBT via the MTS (<u>see Fig. 2, element 100, col. 5, lines 44-67, and col. 6, lines 1-24</u>)." [insertion and underlining for clarity]

Applicant respectfully disagrees. Ramanathan element 100 is not a <u>modem</u> tester and is not adapted to provide a first communication signal to another <u>tester</u> (VBT or voice band tester). Ramanathan element 100 is instead a throughput measurement system as shown in FIG. 2 and explained in Ramanathan col. 5, line 43, through col. 6, line 24:

> "The throughput measurement system 100 evaluates subscriber perceived network performance between the IS 10 and the subscriber sites 12, 14, and 16 and test target 42. ...
>
> <u>The throughput measurement system 100 first selects a target site...</u>the throughput measurement system 100 can emulate a typical data transfer and thereby, measure the network throughput accurately.
>
> ...At the end of the test, the <u>throughput measurement system 100 reports</u> the measured user-perceived <u>throughput as well as the packet loss percentage seen during the test</u>." [deletions and underlining for clarity]

It is respectfully submitted that the claimed limitation of the signals being sent to a tester would not be met even if the modem tester were to read on the non-modem testing throughput measurement system.

The Examiner correctly states that:

> "Ramanathan et al. does not disclose a voice over Internet packet (VoIP) monitoring device, wherein the VoIP monitoring device adapted to monitor the first communication signal, and calculate a first quality of services (QoS) score based on traffic density."

The Examiner continues:

> "However, Murpy et al. discloses a voice over Internet packet (VoIP) monitoring device, wherein the VoIP monitoring device adapted to monitor the first communication signal, and calculate a first quality of services (QoS) score based on traffic density (<u>see Figs. 2 and 3, element 29, col. 4, lines 8-23, and col. 7, lines 32-42</u>)." [underlining for clarity]

Applicant respectfully disagrees. The Applicant's first communication signal comes from a <u>modem tester</u> and the VoIP monitoring device calculates QoS from <u>traffic density</u>. As

shown in Murphy FIGs. 2 and 3, the element 29 is a quality of service monitor existing in a gateway 12 for monitoring signals passing between gateways 12 and 22 calculated on end-to-end delay of the signals. This is confirmed in Murphy col. 4, lines 8-23, which states:

> "A quality of service monitor 29 monitors the QoS of the VoIP network 20. The quality of service monitor 29 is typically VoIP monitoring software that already exists...in the operating system of the gateways." [deletion for clarity]

Murphy col. 7, lines 32-42, states:

> Measuring Quality of Service (QoS) of the VoIP network 20...is determined by the amount of time it takes audio packets to travel between the originating gateway 12 and the destination gateway 22. ...

As would be evident to those having ordinary skill in the art, the traffic density is not necessarily related to travel time. Therefore, the measurement of QoS by Murphy does not render obvious the QoS of the claimed invention.

The Examiner continues:

> "At the time of the invention, it would have been obvious to a person of ordinary skill in the art to combine the VoIP monitoring device teaching by Murpy et al. with Ramanathan et al. The motivation for doing so would have been to provide to monitor for low quality of service condition and reroute the packet which the cost of call is also substantially reduce (col. 2, lines 4-5, 17-19). Therefore, it would have been obvious to combine Murpy et al. and Ramanathan et al. to obtain the invention as specified in the claims 1, 6-10, 14-15, 17, 18, 21, and 25." [underlining for clarity]

Applicant respectfully disagrees. Murphy col. 2, lines 4-5, 17-19, provides a motivation for the invention of Murphy but not for the combination of Ramanathan and Murphy since rerouting packets is detrimental in Ramanathan.

Ramanathan taken as a whole teaches determining network performance by sending limited amounts of data to a target, and counting how many are returned as throughput and how many are lost. Murphy taken as a whole teaches rerouting packets when the return time delay is too long. Ramanathan and Murphy teach different systems which teach away from each other. Further, by rerouting packets in accordance with Murphy, Ramanathan would be rendered inoperative because there would be no throughput for Ramanathan to measure. By limiting the amounts of data transmitted, it is also possible that the end-to-end delay would be artificially altered rendering Murphy inoperative. Thus, either combination would be seemingly inoperative.

In *In re* Gordon, 733 F.2d 900, 902, 221 USPQ 1125, 1127 (Fed. Cir. 1984), the CAFC stated:

> "We have noted elsewhere, as a "useful general rule," that references that teach away cannot serve to create a prima facie case of obviousness... If references taken in combination would produce a "seemingly inoperative device", we have held that such references teach away from the combination and thus cannot serve as predicates for a prima facia case of obviousness." [deletion for clarity]

The Examiner correctly states:

> "Murpy et al. and Ramanathan et al. do not disclose the VBT is adapted to calculate a first transmission impairment test (TIT) score based on the first communication signal and a first received communication signal."

The Examiner continues:

> "However, Jagadeesan discloses the VBT is adapted to calculate a first transmission impairment test (TIT) score based on the first communication signal and a first received communication signal (see Fig. 3, col. 2, lines 58-67, and col. 3, lines 1-11)."

Applicant respectfully disagrees. The untaught limitations of claim related to the VBT include that the VBT receives a first communication signal from a modem tester, the VBT is coupled to a MTS, the TIT score is based on the first communication signal, and the QoS is based on traffic density between the MTS and the VBT.

Jagadeesan is a test system connected to a sound source 10, which generates a series of <u>sound sample</u> frames rather than a <u>communication</u> signal. The Jagadeesan sample and test sound frames are processed to generate parameters combined to calculate a quality score <u>without a TIT score or a QoS score based on traffic density</u>. This is taught in Jagadeesan FIG. 3 and Jagadeesan col. 2, line 58, through col. 3, line 11:

> "A test system 300...is shown in FIG. 3. ...sound source 10 generates the series of sound sample frames x[n]... The signal processor 20 processes the sound sample frames x[n] and outputs...test sound frames y[n]. The series of sound sample frames x[n] and the series of test sound frames y[n] are... [processed to generate] statistical and temporal parameters... The statistical and temporal parameters...[are combined] to calculate an objective sound quality score M.
>
> ...the objective sound quality score values for a number of transmission channels can be analyzed by a selection processor to choose the best transmission channel to carry a voice connection." [deletion for clarity]

Thus, Murphy does not teach or suggest the claimed limitations of the VBT.

The Examiner concludes:

> "At the time of the invention, it would have been obvious to a person of ordinary skill in the art to combine the VBT teaching by Jagadeesan with Murpy et al. and Ramanathan et al. The motivation for doing so would have been to provide to judge the quality of speech and analyze by a selection processor to choose the best transmission channel to carry a voice connection (see col. 3, lines 7-11). Therefore, it would have been obvious to combine Jagadeesan, Murpy et al., and Ramanathan et al. to obtain the invention as specified in the claims 1, 6-10, 14-15, 17, 18, 21, and 25."

Applicant respectfully disagrees. Jagadeesan (see col. 3, lines 7-11), provides a motivation for the invention of Jagadeesan but not for the combination of Jagadeesan with Ramanathan and Murphy. Ramanathan and Murphy are concerned with VoIP packet transmission while Murphy is concerned with voice transmission. As well known to those having ordinary skill in the art, having a good voice connection does not necessary mean that an adequate connection exists for VoIP packets or vice versa.

In re Sang-Su Lee, 277 F.3d 1338, 61 USPQ2d 1430 (Fed. Cir. 2002), the Court held that the conclusion of obviousness may not be made from common knowledge and common sense of a person of ordinary skill in the art without any specific hint or suggestion *in a particular reference*. The specific hint or suggestion is for the combination not for the reason for each invention individually.

Further, it is respectfully submitted that, even assuming for the sake of argument that every statement that the Examiner has made is correct, the conclusions would be incorrect. As explained in *Laitram Corp. v. Cambridge Wire Cloth Co.*, 226 USPQ 298 at 293n (D. Md. Mag. 1985), aff'd in part, rev'd in part, and remanded, 785 F.2d 292, 228 USPQ 935 (Fed. Cir. 1986), cert. denied, 479 U.S. 820 (1986):

> "The question is whether the prior art, considering its scope and content and the level of ordinary skill, must itself suggest the combination of separate elements into the claimed invention in suit, not just whether it illustrates separate elements..."

Based on the above, it is respectfully submitted that claims 1, 6-10, 14-15, 17-18, 21, and 25 are allowable under 35 USC §103(a) as being patentable over Ramanathan in view of Murpy in combination alone, and further in view of Jagadeesan.

Conclusion

In view of the above, Applicant respectfully requests entry of the amended changes. Allowance of claims 1, 3-10, 14, 15, 17-22, and 25 at an early date is solicited.

To the extent necessary, a petition for an extension of time under 37 C.F.R. 1.136 is hereby made. Please charge any shortage in fees due in connection with the filing of this paper, including any extension of time fees, to Deposit Account No. 50-0374 and please credit any excess fees to such deposit account.

Respectfully submitted,

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